Skype for Business 2015 using SIP trunk (TLS) to Cisco Unified Communications Manager Release 10.5.2 SU3
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# Introduction

This document describes the steps and configurations necessary for Cisco Unified Communications Manager (Cisco UCM) release 10.5.2 to interoperate with the Skype for Business 2015 on TLS using the following configuration:

## Cisco UCM Extent and Connect
- Cisco UCM UC service Configuration
- Cisco UCM service Profile Configuration
- Cisco Unified CM IM Presence – CCMCIP Profile Configuration
- Cisco UCM – SIP trunk to Cisco IM&Presence Trunk Configuration
- Cisco UCM end user configuration
- Remote Destination Configuration
- Cisco UCM CTI Remote Device Configuration

## Cisco Unity Connection
- Cisco Unity Connection Telephony Integration – Add Phone System
- Cisco Unity Connection Telephony Integration – Add Port Group
- Cisco Unity Connection Telephony Integration – Add Ports
- Cisco Unity Connection User Configuration

## Acronyms
On the Cisco UCM: PRACK Enabled and Early Offer SIP Profile.

On the Skype for Business: Media Bypass Disabled, Encryption support level Optional

The following items were tested:

- Basic call between the two systems and verification of voice path, using SIP phones on Cisco, and SFB client on the Skype for Business (Refer to limitation section for more info)
- CLIP/CLIR/CNIP/CNIR features: calling party Name and number delivery (allowed and restricted) (Refer to limitation section for more info)
- COLP/CONP/COLR/CONR features: connected Name and number delivery (allowed and restricted) (Refer to limitation section for more info)
- Call transfer: attended and early attended (Refer to limitation section for more info)
- Alerting Name Identification (Refer to limitation section for more info)
- Call forwarding: call forward unconditional(CFU), call forward busy (CFB), and call forward no answer (CFNA)
- Hold and resume with music on hold
- Three-way conferencing (Refer to limitation section for more info)
- Voice messaging and MWI activation-deactivation (Refer to limitation section for more info)
- Extend and Connect (Refer to limitation section for more info)
- Call Park (Refer to limitation section for more info)

Listed below are the highlights of the integration issues:

- Basic calls work from Cisco UCM to Skype for Business and vice versa. Testing has been done with Cisco SIP phones only as SCCP phones do not support 80-bit crypto required by Skype for Business.
- Caller name and number is not updated correctly for basic calls and in the attended and early-attended transfer scenarios.
- Caller ID is updated to “Unknown Number” on Cisco UCM SIP phones in transfer scenarios when a Skype for Business user initiates the transfer.
- Alerting name updates do not occur on Skype for Business.
- Video calls between the Cisco UCM and Skype for Business users were not tested.

Below are the key results:

- Basic call, call transfer, call forwarding, conference call, and hold and resume tested successfully with a few caveats and limitations.
- Centralized voicemail, using Unity Connection server integrated with Cisco UCM via SIP is used for testing. This voicemail solution can provide centralized voicemail services, supporting both Skype for Business and Cisco end-users.
Network Topology

Limitations
These are the known limitations, caveats, or integration issues:

- Skype for Business and Cisco UCM do not support overlap dialing modes on their SIP endpoints.
- Skype for Business does not support alerting name updates.
- Skype for Business does not update the caller ID (connected Name) for a basic or privacy enabled call from the Cisco UCM. Therefore, only the connected party number is displayed on the Cisco UCM Phone.
- Skype for Business does not consider Privacy ID which is being sent by Cisco UCM during 180 Ringing or 200 OK when Connected Name/ID is restricted on Cisco UCM. Consequently, Skype for Business does not update Connected Party display as Private.
- Skype for Business does not update the CLID in transfer/conference scenarios. After the transfer/conference is complete, Cisco UCM sends mid call INVITE and UPDATE messages that contain PAI and RPI. However, Skype for Business does not update this information on its clients.
- In a transfer scenario, when Skype for Business initiates the call transfer, the caller ID of the initial Cisco UCM calling endpoint (transferee) is updated to “Unknown Number” if it is a SIP phone.
- When there is a call leg between two Skype For Business Users, of which either one of the users or both the users are Extend and Connect devices, there is one-way audio. This is noticed during the initiation of some transfer and conference test cases. However when transfer or conference is completed, there is no one-way audio issue noticed and all the parties in the call could hear each other.
• Cisco UCM user is not able to complete an attended transfer when its both the call legs are established with Skype For Business Users that are Extend & Connect devices.

• In a call park scenario, when a Skype for Business client initiates the call park, the Cisco UCM endpoint that retrieves the parked call has its caller ID updated to “Unknown Number”.

• Cisco UCM user parks a call with Skype For Business user. When another Skype For Business user retrieves that call, there is one-way audio. The Skype for Business user, who is retrieved from parking, could not hear the other user.

• Skype for Business does not send PAI by default i.e. when restriction is not enabled. This fails to initiate the Jabber client for call control. The incoming call to a Cisco UCM endpoint is therefore like a regular call without remote destination configuration.
  • This is currently a known issue on the Cisco UCM and is addressed by “CSCuz48313 Tel URI / PAI support in CUCM”.
  • As a workaround, the RD is configured with a “+” prefix and a route pattern to route a DN with a “+’ prefix is also added. (Refer Cisco UCM configuration section - Cisco Unified Communications Manager Route Pattern to invoke Jabber client with Remote Destination configured as Skype for Business Extensions.

• Skype for Business does not support MWI notification from Cisco Unity Connection. It responds with a “405 Method Not Allowed” to a NOTIFY Message from the Cisco UCM that has MWI information.

• In Multiple Call Forwarding scenario between Skype for Business Users and Cisco UCM Users, wherein both originator and terminator being Skype for Business Users, originator does not display the Caller ID of terminator.

• In a call forwarding scenario that involves multiple call forwards and a loop that terminates on a Cisco UCM or Skype for Business user, the calling party (Skype for Business client or Cisco UCM endpoint) hears a re-order tone when it calls the user on which the loop is formed.

• When Cisco UCM user initiates MOH, MOH is RTP. Pad-lock symbol on the phone disappears.

• When Cisco UCM User completes the conference, the audio is on RTP. Pad-lock symbol on the Cisco phone disappears. Audio is on RTP regardless of whether Software based conferencing is used or IOS based conferencing is.

System Components
Hardware Requirements
The following hardware is required:
• Cisco UCS-C240-M3S VMWare Host
• Cisco 7960,7965 ,7975, 9951, and 9971 IP phones
Software Requirements
The following software is required:

- Cisco UCSC-C240-M3S VMware vSphere Image Profile: ESXi-5.5.0-1331820-standard
- Cisco Unified Communications Manager release 10.5.2.13900-12
- Cisco Unified Communications Manager IM & P release 10.5.2.13900-12
- Cisco Unity Connection release 10.5.2.13900-12
- Cisco Jabber 11.6.0 Build 35037
- Skype for Business 2015 6.0.9319.0
- Skype for Business Client version : 15.0.4841.1000
Features
This section lists supported and unsupported features. No deviation from the configuration presented in this document shall be supported by Cisco. Please see the Limitations section for more information.

Features Supported
- CLIP—calling line (number) identification presentation
- CLIR—calling line (number) identification restriction
- CNIP—calling Name identification presentation
- CNIR—calling Name identification restriction
- Alerting Name
- Attended call transfer
- Early attended call transfer
- CFU—call forwarding unconditional
- CFB—call forwarding busy
- CFNA—call forwarding no answer
- COLP—connected line (number) identification presentation
- COLR—connected line (number) identification restriction
- CONP—connected Name identification presentation
- CONR—connected Name identification restriction
- Hold and resume
- Conference call
- MWI—Message Waiting Indicator (only for Cisco Endpoints)
- Audio Codec Preference List
- Call Park/Pickup(see limitation section)
- Extend and Connect
- Shared Line on Cisco Endpoints

Features Not Supported or Not Tested
- Call completion (callback, automatic callback)
- Shared Line on Skype for Business
- Message Waiting Indicator on Skype for Business Endpoints
- Blind transfer
- Video calls
- Scenarios that required 3 PBXs.
- Scenarios involving Non-SIP interfaces.
- Scenarios involving Cisco UCM SCCP Phones
Configuration
The goal of this guide is to provide an overview of the integration between Cisco Unified Communication Manager and Skype for Business. The deployment will interconnect the UC systems using SIP. No PSTN connectivity has been tested with this integration. The following sections provide the required configurations for a successful integration.

Configuring Sequence and Tasks:
Skype for Business:

- Add Cisco UCM to Skype for Business Topology
- Trunk Configuration
- Route Configuration
- Voice Policy and PSTN Usage Configuration
- Dial Plan Configuration
- Call Park range Configuration
- Media Bypass Configuration
- User Configuration
- Client Configuration

Cisco Unified Communications Manager:

- SIP trunk security profile
- SIP profile
- Media resource group and media resource group list
- Assign media resource group list (MRGL) in the default device pool
- Region configuration
- Normalization script
- SIP trunk to Skype for Business
- SIP Trunk to Cisco Unity Connection
- Assign User in Cisco Unity Connection
- SIP and SCCP phones device configuration
- Route Group, Route List and SIP Route Pattern
- Voice Mail
- Route pattern to Skype for Business, Unity Connection and Skype for Business call park range
- Extend and Connect Feature and User configuration

Cisco Unity Connection:

- Cisco Unity Connection Telephony Integration
- Cisco Unity Connection User Configuration
Configuring the Skype for Business

Add Cisco UCM to Skype for Business Topology

Run the Skype for Business 2015 Topology Builder as a user in the CSAdministrator group.

**Navigation:** Skype for Business Server → CleanDefaultTopology → Shared Components → PSTN gateways

Right click and select “New IP/PSTN Gateway”

Set FQDN = <FQDN of the Cisco UCM>—clus20pub.skypelabsj.local is used in this test.

Click Next.

**Skype for Business – Add PSTN Gateway (Continued)**

Check the Enable IPv4 and Use all configured IP addresses radio button

Click Next.
Skype for Business – Add PSTN Gateway (Continued)

Set Trunk Name = FQDN of the Cisco UCM – clus20pub.skypelabsj.local is used for this test
Set Listening port for IP/PSTN gateway = The Listening port should match the Incoming Port setting in the CISCO UCM’s SIP Trunk Security Profile – 5061 is used for this test
Set SIP Transport Protocol = TLS
Set Associate Mediation Server: Assign this PSTN gateway to the Front End co-located mediation server – Medpool.skypelabsj.local is used for this test.
Click Finish.
Publish the topology so these new configurations take effect.
Skype for Business – Add PSTN Gateway (Continued)

Open the Skype for Business 2015 Control Panel.

Navigation: Voice Routing -> Trunk Configuration

Select New ➔ Pool Trunk

![Skype for Business Trunk Configuration](image)
Set Service = Trunk to Cisco UCM that was created earlier as a PSTN gateway in the topology builder – clus20pub.skypelabsj.local is used for the test.

Set Maximum early dialogs supported = 20
Set Encryption support level = Optional
Set Refer Support = Enable sending refer to the gateway
Uncheck Enable media bypass
Check Centralized media processing
Uncheck Enable RTP latching
Check Enable forward call history
Uncheck Enable forward P-Asserted-Identity data* [Note: this is checked when test scenarios that involve restrict ID need to be executed]
Uncheck Enable outbound routing failover timer
Skype for Business – Trunk Configuration (Continued)

Edit Trunk Configuration - PstnGateway:clus20pub.skypelabsj.local

- Scope: Pool
- Name: PstnGateway:clus20pub.skypelabsj.local
- Description:
- Maximum early dialogs supported: 20
- Encryption support level: Optional
- Refer support: Enable sending refer to the gateway
- Enable media bypass
- Centralized media processing
- Enable RTP latching
- Enable forward call history
- Enable forward P-Asserted-Identity data
- Enable outbound routing failover timer
Skype for Business – Trunk Configuration (Continued)

Add a Translation rule under Called number translation rules – CUCMExtn was created in this test. This is used to remove the “+” that is added by Skype for Business during a transfer to a Cisco UCM extension. If Skype for Business attempts a transfer to a Cisco UCM extension without this rule, the transfer fails because the extension is not recognized by Cisco UCM.
Skype for Business –Trunk Configuration- Translation Rule

Edit Trunk Configuration  Edit Called Number Translation Rule - CUCMExtn

Name: *  CUCMExtn

Description:  

Build a Translation Rule

Fill in the fields that you want to use, or create the rule manually by clicking Edit.

Starting digits:
+  

Length:
At least ▼ 2 ▼

Digits to remove:
1 ▼

Digits to add:

Pattern to match: *
\^\+(\d+)\$

Translation rule: *
$1
Skype for Business – Trunk Configuration (Continued)

Skype for Business – Trunk Configuration

Navigation: Voice Routing -> Route

Click New

Set Name = enter a name to identify this Route. SFB-Cisco is used for this test.

Add Associated trunks = select the trunk configured earlier – PstnGateway:clus20pub.skypelabsj.local
Create voice routing test case information

Edit Voice Route - SFB-Cisco

Scope:
Name: *
SFB-Cisco
Description:

Build a Pattern to Match
Add the starting digits that you want this route to handle, or create the expression manually by clicking Edit.
Starting digits for numbers that you want to allow:
Type a valid number and then click Add.

Match this pattern: *
.

 Suppress caller ID
Alternate caller ID:
Skype for Business Voice Policy and PSTN Usage Configuration

**Navigation**: Voice Routing -> Voice Policy

Click New

Set Name = enter a name to identify this voice policy – SFB-Cisco is used in this test.

Set Calling Features:

- Check Enable call forwarding
- Check Enable delegation
- Check Enable call transfer
- Check Enable call park
- Check Enable simultaneous ringing of phones
- Check Enable team call
- Check Enable PSTN reroute
- Uncheck Enable bandwidth policy override
- Uncheck Enable malicious call tracing

Set Associated PSTN usages:

- Click New

- Set Name: enter a name to identify this PSTN Usage record – SFB_PSTN is used in the test.

- Set Associated Routes = select the route created earlier= SFB-Cisco
Skype for Business Dial Plan Configuration

**Navigation:** Voice Routing-> Dial Plan

Add a new User dial plan and a new Pool dial plan.

User dial plan:

Set Name – enter text to identify this dial plan – *cucm* is used in this test.

A user dial plan with a normalization rule was configured for this test:

- **CUCM 4 Digit:** To reach the 4 digit extensions from Cisco UCM – This allows 4 digits to be dialed and not undergo any normalization.
Pool dial plan:

Select Service: PstnGateway:clus20pub.skypelabsj.local is selected

Set Simple Name= enter text to identify this pool dial plan. PstnGateway_clus20pub.skypelabsj.local is used in this test

Associated Normalization Rules→New

Set Name: enter text to identify this rule – Call pick up From CUCM was created in this test
This is to accept the call park range dialed by Cisco UCM users to retrieve a call parked by the Skype for Business client.

**Skype for Business – Pool Dial Plan-Normalization Rule 1**

Add another normalization rule as below:

This is used by the client to dial out to internal extensions and to the external PBX i.e. Cisco UCM.
Skype for Business – Pool Dial Plan-Normalization Rule 2

Edit Dial Plan  Edit Normalization Rule - Keep All

Name: * Keep All

Description:

Build a Normalization Rule
Fill in the fields that you want to use, or create the rule manually by clicking Edit.

Starting digits:

Length:
At least 1

Digits to remove:

Digits to add:

Pattern to match: *
^\(d+)$

Translation rule: *
$1$

OK  Cancel

Internal extension
Skype for Business Call Park Range Configuration

**Navigation:** Voice Features -> Call Park

Click New.

Set Name = enter text to identify this call park range – Orbit range is used in the test.

Set Number Range = 100 to 199 is used in the test.

Set FQDN of destination server= select the desired server - FE01.skypelabsj.local is used in the test
Skype for Business Global Media Bypass Configuration

**Navigation:** Network Configuration -> Global

Edit Global Setting –

- Uncheck Enable media bypass

Commit the configuration.
Skype for Business User Configuration
Login to the Skype for Business Active Directory

Navigation: Active Directory Users and Computers → Users

Add a New User
Skype for Business – New User configuration (continued)

Follow the screenshots below to add a new user:

![New Object - User window](image-url)

- **First name:** SFBUser1
- **Initials:** [Blank]
- **Last name:** [Blank]
- **Full name:** SFBUser1

**User login name:**
- **user1**
- **@skypelabsj.local**

**User login name (pre-Windows 2000):**
- **SKYPELABSJ\user1**

[Back] [Next] [Cancel]
Skype for Business – New User configuration (continued)

New Object - User

Create in: skypelabj.local/Users

Password: ************
Confirm password: ************

- User must change password at next logon
- User cannot change password
- Password never expires
- Account is disabled

< Back  Next >  Cancel
Once the user is created, login to the Skype for Business 2015 Control Panel
Navigation: Users → Enable users

Click on the Add button and find the new user created earlier.
Set Assign users to a pool= FE01.skypelabsj.local from drop down menu
Set Generate user’s SIP URI: Specify a SIP URI: sip:SFBUser1@skypelabsj.local. This is used in this test
Set Telephony=Enterprise Voice
Set Line URI: = tel:+8003 is used for the test. This is the DN for the user.
Set Dial plan policy = cucm (as configured earlier)
Set Voice policy= SFB-Cisco (as configured earlier)
Click Enable.
Skype for Business – New User configuration (continued)

Download the latest version of the Skype for Business client and launch the same.

Navigation: Settings → Tools → Options → Personal → MyAccount

Set Sign-in-address= enter the sip uri of the user configured in username@domain format.  
sfbuser1@skypelabsj.local is used for example.
Click Advanced. Select Manual Configuration.

Set Internal Server Name= Enter the FQDN of the domain (skypelabsj.local is used for example)

**Skype for Business – Client configuration (continued)**
Cisco Unified Communications Manager SIP Trunk Security Profile for Trunk to Skype for Business

**Navigation**: System → Security → SIP trunk security profile

Set Name* = SFB-secure. This is used for the test.

Set Device Security mode = Encrypted

Set Incoming Transport Type = TLS

Set Outgoing Transport Type = TLS

Set X.509 Subject Name = medpool.skypelabsj.local (medpool.skypelabsj.local is the Subject Name of the SIP certificate created in Mediation Server)

Incoming Port = 5061

Check Accept Presence Subscription

Uncheck Accept out of dialog refer

Check Accept unsolicited notification

Check Accept Replaces header

Check Transmit security status
Cisco Unified Communications Manager SIP Trunk Security Profile for Trunk to Unity Connection

**Navigation:** System → Security → SIP trunk security profile

Set Name* = TLS_CUC. This is used for the test.

Set Device Security mode = Encrypted

Set Incoming Transport Type = TLS

Set Outgoing Transport Type = TLS

Set X.509 Subject Name = TLSConnection (TLSConnection is the Subject Name of the SIP certificate created in Cisco Unity Connection)

Incoming Port = 5061

Unchecked Accept Presence Subscription

Check Accept out-of-dialog refer**

Check Accept unsolicited notification
Check Accept Replaces header
Check Transmit security status
All other values are default.

Cisco Unified Communications Manager SIP Profile

**Navigation:** Device → Device Settings → SIP Profile

Set Name* = SFB - Standard SIP Profile. This is used for this test.
Set Description = this text is used to identify this SIP Profile.
Set SIP Rel1XX Options = Send PRACK if 1xx Contains SDP
Set Early Offer support for voice and video calls = Best Effort (no MTP inserted)
Check Enable OPTIONS Ping to monitor Destination status for Trunks with Service Type "None (Default)"

All other values are default.
Status

- Status: Ready
- All SIP devices using this profile must be restarted before any changes will take affect.

SIP Profile Information

- Name: SFB - Standard SIP Profile
- Description: Default SIP Profile
- Default MTP Telephony Event Payload Type: 101
- Early Offer for G. Clear Calls: Disabled
- User-Agent and Server header information: Send Unified CM Version Information as User-Agent
- Version in User Agent and Server Header: Major And Minor
- Dial String Interpretation: Phone number consists of characters 0-9, *, #, and v
- Confidential Access Level Headers: Disabled
- Redirect by Application
- Disable Early Media on 100
- Outgoing T.38 INVITE include audio mime
- Use Fully Qualified Domain Name in SIP Requests
- Assured Services SIP conformance
Cisco Unified Communications Manager SIP Profile (Continued)

SIP Profile Configuration

- SDP Information
  - SDP Session-level Bandwidth Modifier for Early Offer and Re-invites: TIAS and AS
  - SDP Transparency Profile: Pass all unknown SDP attributes
  - Accept Audio Codec Preferences in Received Offer
    - Require SDP Inactive Exchange for Mid-Call Media Change: Default
  - Allow RA/RS bandwidth modifier (RFC 3556)

- Parameters used in Phone
  - Timer Invite Expires (seconds): 180
  - Timer Register Delta (seconds): 5
  - Timer Register Expires (seconds): 3600
  - Timer T1 (msec): 500
  - Timer T2 (msec): 4000
  - Retry INVITE: 6
  - Retry Non-INVITE: 10
  - Start Media Port: 16384
  - Stop Media Port: 32766
  - Call Pickup URI: x-cisco-service-url-pickup
  - Call Pickup Group Other URI: x-cisco-service-url-pickup
  - Call Pickup Group URI: x-cisco-service-url-pickup

Cisco Unified Communications Manager SIP Profile (Continued)

SIP Profile Configuration

- Call Pickup Group URI: x-cisco-service-url-pickup
- Meet Me Service URI: x-cisco-service-url-meetme
- User Info: None
- DTMF DB Level: Nominal
- Call Hold Ring Back: Off
- Anonymous Call Block: Off
- Caller ID Blocking: Off
- Do Not Disturb Control: User
- Telnet Level for 7940 and 7960: Disabled
- Resource Priority Namespace: < None >
- Timer Keep Alive Expires (seconds): 120
- Timer Subscribe Expires (seconds): 120
- Timer Subscribe Delta (seconds): 5
- Maximum Redirects: 70
- Off Hook To First Digit Timer (milliseconds): 15000
- Call Forward URL: x-cisco-service-url-cfwdall
- Speed Dial (Abbreviated Dial) URI: x-cisco-service-url-abbreviated
- Conference Join Enabled
- RFC 2543 Hold
- Semi-Attended Transfer
Cisco Unified Communications Manager SIP Profile (Continued)
Cisco Unified Communications Manager SIP Profile (Continued)

**Media Resource Group MRG**

Set Name* = MRG, This is used for this test.

Set Description = this text is used to identify this Media Resource Group.

Set all resources in the Selected Media Resources* Box.

All other values are default.
Resource Group for MRG_NoMTP

Set Name* = MRG_NoMTP. This is used for the test.

Set Description = this text is used to identify this Media Resource Group.

Set Available Media Resources = MTP_2, MTP_3 and MTP_4

Set other resources in the Selected Media Resources*

All other values are default.
Cisco Unified Communications Manager Media Resource Group Configuration

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<th>Related Links: Back To Find/List</th>
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<tr>
<td>Status</td>
<td>Status: Ready</td>
<td></td>
</tr>
<tr>
<td>Media Resource Group Status</td>
<td>Media Resource Group: MRG_NoMTP (used by 26 devices)</td>
<td></td>
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<tr>
<td>Media Resource Group Information</td>
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<td></td>
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<tr>
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<td>MRG_NoMTP</td>
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<td>Selected Media Resources</td>
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<td>ANN_4 (ANN)</td>
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<td>CFB_2 (CFB)</td>
<td></td>
</tr>
<tr>
<td></td>
<td>CFB_3 (CFB)</td>
<td></td>
</tr>
<tr>
<td>Use Multi-cast for MCH Audio (if at least one multi-cast MCH resource is available)</td>
<td></td>
<td></td>
</tr>
</tbody>
</table>

Cisco Unified Communications Manager Media Resource Group Configuration

Find and List Media Resource Groups

<table>
<thead>
<tr>
<th>Status</th>
<th>2 records found</th>
</tr>
</thead>
</table>

<table>
<thead>
<tr>
<th>Media Resource Group (1 - 2 of 2)</th>
<th>Rows per Page 50</th>
</tr>
</thead>
<tbody>
<tr>
<td>Find Media Resource Group where Name begins with</td>
<td>Find</td>
</tr>
<tr>
<td>MRG</td>
<td>false</td>
</tr>
<tr>
<td>MRG_NoMTP</td>
<td>false</td>
</tr>
</tbody>
</table>

Add New | Select All | Clear All | Delete Selected
Add New

Set Name*= MRGL. This is used for this test.

Set Description = this text is used to identify this Media Resource Group List.

Set Available Media Resources = MRGL

Set Selected Media Resource Groups= MRG

Add new

Set Name*= MRGL_noMTP. This is used for the test

Set Description = this text is used to identify this Media Resource Group List

Set Available Media Resources MRG

Set Selected Media Resource Groups= MRGL_NoMTP
Cisco Unified Communications Manager Media Resource Group List Configuration

Find and List Media Resource Group Lists

<table>
<thead>
<tr>
<th>Status</th>
</tr>
</thead>
<tbody>
<tr>
<td>2 records found</td>
</tr>
</tbody>
</table>

<table>
<thead>
<tr>
<th>Media Resource Group List (1 – 2 of 2)</th>
<th>Rows per Page 50</th>
</tr>
</thead>
<tbody>
<tr>
<td>Find Media Resource Group List where Name begins with</td>
<td>Find</td>
</tr>
<tr>
<td>MRGL</td>
<td></td>
</tr>
<tr>
<td>MRGL NoMTP</td>
<td></td>
</tr>
</tbody>
</table>

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Cisco Unified Communications Manager Device Pool Configuration

Device Pool - **G711 Preferred** is created in this test.

**Navigation Path:** System → Device Pool

Add New.

Set Device Pool Name* = G711 Preferred. This is used in the test.

Set Cisco Unified Communications Manager Group* = Default

Set Date/Time Group* = CMLocal

Set Region* = G711 Preferred. This is used in this example

Set Media Resource Group List = MRGL. This is used in this example.

All other values are default.
Cisco Unified Communications Manager Region Configuration

**Navigation Path:** System → Region Information → Region

Add New

G711 Preferred is created in this test.

Set Name *= G711 Preferred. This is used in this example

Set Region = G711 Preferred. This is used in this example

Set Audio Codec Preference List = G711 Preferred

Set Maximum Audio Bit Rate = 64 Kbps (G7.22, G7.11). This is used in this example

Set Region = Default. This is used in this example

Set Audio Codec Preference List = G711 G729. This is used in this example

Set Maximum Audio Bit Rate = 64 Kbps (G722, G7.11). This is used in this example

All other values are default

---

Cisco Unified Communications Manager Normalization Script

**Navigation:** Device->Device Settings->SIP Normalization Script

Add New

Set Name = enter text here to identify the normalization script for use on trunk. lync_interop_updated is used in this test.

Set Content = add script content.

**Note:** “lync_interop” was the originally provided script by Cisco Support for Cisco UCM-Skype for Business TLS integration. However, with the script activated, the call from Skype for Business to Cisco UCM was not established on SRTP still. Cisco UCM sent AVP (that is, chose RTP instead of SRTP) when Skype for Business sent a=tcap:1 RTP/SAVP through INVITE SDP. The script was updated to fix this issue and “lync_interop_updated” is the updated script.
Normalization Script

```---

Description:
Provides interoperability for Microsoft Lync

Handle Below Scenarios

1. Add user=phone for all outbound Invite messages because it is mandatory for Lync

2. Change the CT=Line values to 1000, Moderate bandwidth in all outgoing messages from CUCM to Lync

3. There is Remote ringback hear issue
There is issue with PRACK enabled on CUCM and media bypass enabled on Lync. Enabling media bypass on Lync allows the rtp from lync endpoint to flow through CUCM directly instead of flowing through mediation server. The problem with PRACK enabled is that Lync end
```
point is now not able to answer the incoming call. Looking into the traces, it appears that even though Lync sent updated connection information in 183 w/sdp, the call manager is still sending rtp to the mediation server which seems to be incorrect. So in this scenario CUCM expects 180 Ringing not 183 Session progress. So added the Script to convert 183 Session Progress to 180 Ringing.

4. There is incoming Invite from Lync and in From Header there is "user=phone" which cause CUCM to send malformed data in to different layers which cause call failure. So this is work around for that scenario.

5. Script modify the AS header which from outgoing messages because call forward fails due to bandwith negotiation value is A=64 is not supported

6. Script convert the History info to diversion Header since call forward to unity is not supported.

7. Transfer Scenario: Referred-By in Incoming Invite is converted to Diversion Header.

Script Parameters:

Release: 9.1(2), 10.0.(1)

Copyright (c) 2009-2011 Cisco Systems, Inc. All rights reserved.
local function modify_CT_bandwidth(msg)
    local sdp = msg:getSdp()
    if sdp
        then
            local b_CT_line = sdp:getLine("b=CT","64")
            if not b_CT_line
                then
                    local b_CT_line = sdp:getLine("b=CT","0")
                    if not b_CT_line
                        then
                            return
                        end
                    b_CT_line = b_CT_line:gsub("0", "1000")
                    sdp = sdp:modifyLine("b=CT", "0", b_CT_line)
                    msg:setSdp(sdp)
                    return
                end
            b_CT_line = b_CT_line:gsub("64", "1000")
            sdp = sdp:modifyLine("b=CT", "64", b_CT_line)
            msg:setSdp(sdp)
        end
    end
end

local function remove_AS_bandwidth(msg)
    local sdp = msg:getSdp()
    if sdp
        then
            local b_AS_line = sdp:getLine("b=AS","64")
            if b_AS_line
                then
                    sdp = sdp:removeLine("b=AS", "64")
                    msg:setSdp(sdp)
                end
        end
end

local function process_outbound_request(msg)
    local method, ruri, ver = msg:getRequestLine()
    if string.find(ruri, "@")
        then
            local uri = ruri .. ";user=phone"
            msg:setRequestUri(uri)
end
modify_CT_bandwidth(msg)
remove_AS_bandwidth(msg)
end

local function process_outbound_message(msg)
modify_CT_bandwidth(msg)
remove_AS_bandwidth(msg)
end

local function process_inbound_progress(msg)

msg:setResponseCode(180, "Ringing")
local sdpt = msg:getSdp()

if sdpt then
    sdpt = sdpt:removeMediaDescription("audio")
    msg:setSdp(sdp)
end

local req = msg:getHeader("Require")
local reqHeader = req
if req
    msg:removeHeader("Require")
end

local rseq = msg:getHeader("Rseq")
local rseqPresnt = rseq
if rseq
    seqVal = msg:getHeaderValues("Rseq")
    msg:removeHeader("Rseq")
end

local sdpt = msg:getSdp()
if sdpt
    msg:removeUnreliableSdp()
end

if reqHeader
    msg:addHeader("Require", "100rel")
end
if rseqPresnt
then
    msg:addHeader("RSeq",seqVal[1])
end
end

-- Future reference for changing cause values in diversion header scenario
-- local HiCauseToDiversion = {}
-- HiCauseToDiversion["302"] = "unconditional"
-- HiCauseToDiversion["486"] = "user-busy"
-- HiCauseToDiversion["408"] = "no-answer"
-- HiCauseToDiversion["480"] = "deflection"
-- HiCauseToDiversion["487"] = "deflection"
-- HiCauseToDiversion["503"] = "unavailable"
-- HiCauseToDiversion["404"] = "unknown"

function convertHIToDiversion(msg)
    local historyInfos = msg:getHeaderValues("History-Info")
    for i, hi in ipairs(historyInfos) do
        hi = string.gsub(hi, "%%3B", ";")
        hi = string.gsub(hi, "%%3D", "=")
        hi = string.gsub(hi, "%%22", "\"")
        hi = string.gsub(hi, "%%20", " ")

        -- MS format: <sip:+19728522619@med02.lynclabsj.local;user=phone>;index=1;ms-retarget-reason=forwarding
        local uri, index, reason = string.match(hi, "<\(sip:.*@\.).*\);index=(.*)reason=(.*)")
        trace.format("hi: uri '%s', reason '%s'", uri or "nil", reason or "nil")

        if uri then
            local diversion = string.format("<%s>", uri)
            if reason then
                diversion = string.format("<%s>;reason="/unconditional"", uri)
            end
            msg:addHeader("Diversion", diversion)
        end
    end
end

function convertReferredByToDiversion(msg)
local refInfo = msg:getHeader("Referred-By")
if refInfo
    then
        local diversion = string.format("%s;reason="unconditional", refInfo)
        msg:addHeader("Diversion", diversion)
    end
end

local function replaceHistoryHeader(msg)
    local hist = msg:getHeader("History-Info")
    if hist
        then
            convertHIToDiversion(msg)
            local di = msg:getHeader("Diversion")
            if di
                then
                    msg:removeHeader("History-Info")
                end
        end
    end
end

local function replaceReferredByHeader(msg)
    local refby = msg:getHeader("Referred-By")
    if refby
        then
            convertReferredByToDiversion(msg)
        end
end

local function modifyUserFrom(msg)
    -- get a data from "From" header and replace
    local removeUser = ""
    local value = msg:getHeader("From")
    if value
        then
            value = value:gsub(";user="phone", removeUser)
            if value
                then
local function process_inbound_request(msg)
    modifyUserFrom(msg)
    replaceHistoryHeader(msg)
    replaceReferredByHeader(msg)
    removecryptoline(msg)
    local sdp = msg:getSdp()
    if sdp then
        local tcap = sdp:getLine("a=tcap:", "RTP/SAVP")
        if tcap then
            local a_m_line = sdp:getLine("m=audio", "RTP/AVP")
            a_m_line = a_m_line:gsub("AVP", "SAVP")
            sdp = sdp:modifyLine("m=audio", "RTP/AVP", a_m_line)
        end
        sdp = sdp:removeLine("a=crypto:", |2^31|)
        msg:setSdp(sdp)
    end
end

function process_inbound_any_response(msg)
    msg:addHeader("SUPPORTED","X-cisco-srtp-fallback")
    local sdp = msg:getSdp()
    if sdp then
        trace.format("Inbound SDP")
        local tcap = sdp:getLine("a=tcap:", "RTP/SAVP")
        if tcap then
            local a_m_line = sdp:getLine("m=audio", "RTP/AVP")
            a_m_line = a_m_line:gsub("AVP", "SAVP")
            sdp = sdp:modifyLine("m=audio", "RTP/AVP", a_m_line)
        end
        sdp = sdp:removeLine("a=crypto:", |2^31|)
        trace.format("after removing crypto")
        msg:setSdp(sdp)
    end
end
function removecryptoline(msg)
    local sdp = msg:getSdp()
    if sdp then
        trace.format("removecryptoline before removing crypto")
        sdp = sdp:removeLine("a=crypto:","\^31")
        trace.format("removecryptoline after removing crypto")
        msg:setSdp(sdp)
    end
end

function process_inbound_any_request(msg)
    msg:addHeader("SUPPORTED","X-cisco-srtp-fallback")
end

M.outbound_INVITE = process_outbound_request
M.outbound_ACK = process_outbound_message
M.outbound_200_INVITE = process_outbound_message
M.outbound_18X_INVITE = process_outbound_message
M.inbound_183_INVITE = process_inbound_progress
M.inbound_INVITE = process_inbound_request
M.inbound_ANY_ANY = process_inbound_any_response
M.inbound_ANY = process_inbound_any_request

return M

Cisco Unified Communications Manager SIP Trunk to Skype for Business Configuration

**Navigation:** Device → Trunk

Set Device Name* = SFB-MedPool-CUCM. This is used for the test

Set Description = this text is used to identify this Trunk Group

Set Device Pool* = G711 Preferred. This is used for the test

Set Call Classification* = Use System Default. This is used for the test

Set Media Resource Group List = MRGL. This is used for the test

Uncheck Media Termination Point Required

Check Run On All Active Unified CM Nodes

Check Redirecting Diversion Header Delivery – Inbound
Set Destination Address = medpool.skypelabsj.local. [FQDN of Skype for Business Mediaition Pool] This is used in the test

Set SIP Trunk Security Profile* = SFB-secure

Set SIP Profile* = SFB – Standard SIP Profile

Set DTMF Signaling Method* = RFC 2833

Set Normalization Script = lync_interop_updated

All other values are default.
Cisco Unified Communications Manager SIP Trunk to Skype for Business Configuration (Continued)

<table>
<thead>
<tr>
<th>Incoming Called Party Settings</th>
<th></th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Number Type</strong></td>
<td><strong>Prefix</strong></td>
</tr>
<tr>
<td>Incoming Number</td>
<td>Default</td>
</tr>
</tbody>
</table>

<table>
<thead>
<tr>
<th>Connected Party Settings</th>
<th></th>
</tr>
</thead>
<tbody>
<tr>
<td>Connected Party Transformation CSS</td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>Use Device Pool Connected Party Transformation CSS</td>
<td></td>
</tr>
</tbody>
</table>

<table>
<thead>
<tr>
<th>Outbound Calls</th>
<th></th>
</tr>
</thead>
<tbody>
<tr>
<td>Called Party Transformation CSS</td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>Use Device Pool Called Party Transformation CSS</td>
<td></td>
</tr>
<tr>
<td>Calling Party Transformation CSS</td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>Use Device Pool Calling Party Transformation CSS</td>
<td></td>
</tr>
<tr>
<td>Calling Party Selection</td>
<td></td>
</tr>
<tr>
<td>Calling Line ID Presentation</td>
<td></td>
</tr>
<tr>
<td>Calling Name Presentation</td>
<td></td>
</tr>
<tr>
<td>Calling and Connected Party Info Format</td>
<td></td>
</tr>
<tr>
<td>Redirecting Division Header Delivery - Outbound</td>
<td></td>
</tr>
<tr>
<td>Redirecting Party Transformation CSS</td>
<td>&lt; None &gt;</td>
</tr>
</tbody>
</table>
Cisco Unified Communications Manager SIP Trunk to Skype for Business Configuration (Continued)

Trunk Configuration

SIP Information

Destination

- Destination Address is an SRV

Destination Address | Destination Address IPv6 | Destination Port
-------------------|-------------------------|-------------------
medpool.skype.com,local |  | 5061

MTP Preferred Originating Codec
BLF Presence Group
SIP Trunk Security Profile
Routing Calling Search Space
Out-Of-Dialing Refer Calling Search Space
SUBSCRIBE Calling Search Space
SIP Profile
DTMF Signaling Method

Cisco Unified Communications Manager SIP Trunk to Skype for Business Configuration (Continued)

Normalization Script

Normalization Script: lyncinterop_updated

Enable Trace

Parameter Name | Parameter Value
---------------|------------------

Recording Information

- None
- This trunk connects to a recording-enabled gateway
- This trunk connects to other clusters with recording-enabled gateways

Geolocation Configuration

Geolocation | Geolocation Filter
< None > | < None >

Send Geolocation Information

Save | Delete | Reset | Add New

- Indicates required item.
** Indicates a device reset is not required for changes to Packet Capture Mode and Packet Capture Duration.

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Cisco Unified Communications Manager SIP Trunk to Cisco Unity Connection Configuration

**Navigation**: Device → Trunk

Set Device Name*= UnityConnection-TLS. This is used for the test.
Set Description = this text is used to identify this Trunk Group.
Set Device Pool* = G711 Preferred

Check Run On All Active Unified CM Nodes
Check Redirecting Diversion Header Delivery – Inbound
Check Redirecting Diversion Header Delivery – Outbound
Set Destination Address = 10.80.10.5. This is used for the test.

Set SIP Trunk Security Profile*= TLS_CUC
Set SIP Profile*= SFB-Standard SIP Profile
DTMF Signaling Method *= RFC 2833
Set Normalization Script = lync_interop_updated

All other values are default
### Device Information

<table>
<thead>
<tr>
<th>Product</th>
<th>SIP Trunk</th>
</tr>
</thead>
<tbody>
<tr>
<td>Device Protocol</td>
<td>SIP</td>
</tr>
<tr>
<td>Trunk Service Type</td>
<td>None (Default)</td>
</tr>
<tr>
<td>Device Name</td>
<td>UnityConnection-TLS</td>
</tr>
<tr>
<td>Description</td>
<td>UnityConnection Secure</td>
</tr>
<tr>
<td>Device Role</td>
<td>G711 Preferred</td>
</tr>
<tr>
<td>Common Device Configuration</td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>Cell Classification</td>
<td>Use System Default</td>
</tr>
<tr>
<td>Media Resource Group List</td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>Location</td>
<td>Hub_None</td>
</tr>
<tr>
<td>AAR Group</td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>Tunneled Protocol</td>
<td>None</td>
</tr>
<tr>
<td>QoS10 Version</td>
<td>No Changes</td>
</tr>
<tr>
<td>ASK1 ROSE OID Encoding</td>
<td>No Changes</td>
</tr>
<tr>
<td>Packet Capture Mode</td>
<td>None</td>
</tr>
<tr>
<td>Packet Capture Duration</td>
<td>0</td>
</tr>
</tbody>
</table>
Cisco Unified Communications Manager SIP Trunk to Cisco Unity Connection Configuration (Continued)

<table>
<thead>
<tr>
<th>Trunk Configuration</th>
<th>Related Links: Back To Find/List</th>
<th>Go</th>
</tr>
</thead>
<tbody>
<tr>
<td>Save</td>
<td>Delete</td>
<td>Refresh</td>
</tr>
<tr>
<td>Media Termination Point Required</td>
<td></td>
<td></td>
</tr>
<tr>
<td>Retry Video Call as Audio</td>
<td></td>
<td></td>
</tr>
<tr>
<td>Path Replacement Support</td>
<td></td>
<td></td>
</tr>
<tr>
<td>Transmit UTP-B for Calling Party Name</td>
<td></td>
<td></td>
</tr>
<tr>
<td>Transmit UTP-B Names in QSIG AODU</td>
<td></td>
<td></td>
</tr>
<tr>
<td>Unattended Port</td>
<td></td>
<td></td>
</tr>
<tr>
<td>SRTP Allowed* - When this flag is checked, encrypted TLS needs to be configured in the network to provide end-to-end security. Failure to do so will expose keys and other information.</td>
<td></td>
<td></td>
</tr>
<tr>
<td>Consider Traffic on This Trunk Secure*</td>
<td></td>
<td></td>
</tr>
<tr>
<td>Route Class Signaling Enabled*</td>
<td></td>
<td></td>
</tr>
<tr>
<td>Use Trusted Relay Point*</td>
<td></td>
<td></td>
</tr>
<tr>
<td>PSTN Access</td>
<td></td>
<td></td>
</tr>
<tr>
<td>Run On All Active Unified CM Nodes</td>
<td></td>
<td></td>
</tr>
</tbody>
</table>

<table>
<thead>
<tr>
<th>Intercompany Media Engine (IME)</th>
<th></th>
</tr>
</thead>
<tbody>
<tr>
<td>E.164 Transformation Profile</td>
<td>None &gt;</td>
</tr>
</tbody>
</table>

<table>
<thead>
<tr>
<th>MLPP and Confidential Access Level Information</th>
<th></th>
</tr>
</thead>
<tbody>
<tr>
<td>MLPP Domain</td>
<td>None &gt;</td>
</tr>
<tr>
<td>Confidential Access Mode</td>
<td>None &gt;</td>
</tr>
<tr>
<td>Confidential Access Level</td>
<td>None &gt;</td>
</tr>
</tbody>
</table>
Cisco Unified Communications Manager SIP Trunk to Cisco Unity Connection Configuration (Continued)
Cisco Unified Communications Manager SIP Trunk to Cisco Unity Connection Configuration (Continued)
Cisco Unified Communications Manager SIP Trunk to Cisco Unity Connection Configuration
(Continued)

- **Trunk Configuration**

- **Caller Information**
  - Caller ID DN
  - Caller Name
  - Maintain Original Caller ID DN and Caller Name in Identity Headers

- **SIP Information**
  - Destination Address is an SRV
  - Destination Address: 10.80.10.5
  - Destination Address IPv6
  - Destination Port: 5061
  - MTP Preferred Originating Codec
  - BLF Presence Group
  - SIP Trunk Security Profile: TLS CUC
  - Re-routing Calling Search Space: < None >
  - Out-Of-Dial Calling Search Space: < None >
  - SUBSCRIBE Calling Search Space: < None >
  - SIP Profile: Standard SIP Profile
  - DTMF Signaling Method: RFC 2833

- **Normalization Script**
  - Normalization Script: tync_interop_updated
  - Enable Trace
  - Parameter Name: 
  - Parameter Value:

- **Recording Information**
  - None
  - This trunk connects to a recording-enabled gateway
  - This trunk connects to other clusters with recording-enabled gateways

- **Geolocation Configuration**
  - Geolocation: < None >
  - Geolocation Filter: < None >
  - Send Geolocation Information

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Cisco Unified Communications Manager Route Group

Navigation: Call Routing → Route/Hunt → Route Group

Add New

SFB-CUCM was configured in this test

Set Route Group Name = SFB-CUCM

Set Distribution Algorithm = Circular

Select SFB-MedPool-CUCM from Available Devices and click the Add to Route Group
Cisco Unified Communications Manager Route List

**Navigation:** Call Routing → Route/Hunt → Route List

Add New

SFB-CUCM_Route List was created for this test.

Set Name: SFB-CUCM_Route List

Set Cisco Unified Communications Manager Group = Default

Click on Add Route Group

Set Route Group* = SFB-CUCM-[NON-QSIG]
Cisco Unified Communications Manager SIP Route Pattern

**Navigation:** Call Routing → SIP Route Pattern

Add New

Set IPv4 Pattern* = medpool.skypelabsj.local. This is the FQDN of the Skype for Business Front End server.

Set SIP Trunk/Route List* = SFB_CUCM_Route List

In a similar way, add SIP Route Patterns for all the servers that comprise the Skype for Business environment.

In the test, the following SIP Route Patterns were configured:
Cisco Unified Communications Manager Voice Mail Configuration

Configure Voice Mail Pilot:

Navigation: Advanced Features → Voice Mail → Voice Mail Pilot

Add new

Set Voice Mail Pilot Number = 7000. This is used for the test

Set Description = Unity Connection VM. This text is used to identify this SIP Profile

---

Cisco Unified Communications Manager Route Pattern to Skype for Business Extensions

**Navigation:** Call Routing → Route/Hunt → Route Pattern

Add New

Set Route Pattern* = 8XXX. This is used to route to the Skype for Business in this test

Set Description = this text is used to identify this Route Pattern

Set Gateway/Route List* = SFB-CUCM_Route List. This is used for the test

Uncheck Provide Outside Dial Tone

Set Calling Line ID Presentation= Default

Set Calling Name Presentation= Default

Set Connected Line ID Presentation* = Default

Set Calling Name Presentation* = Default

All other values are default.
## Route Pattern Configuration

<table>
<thead>
<tr>
<th>Field</th>
<th>Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>Route Pattern</td>
<td>8XXX</td>
</tr>
<tr>
<td>Route Partition</td>
<td>&lt;None&gt;</td>
</tr>
<tr>
<td>Description</td>
<td></td>
</tr>
<tr>
<td>Numbering Plan</td>
<td>-- Not Selected --</td>
</tr>
<tr>
<td>Route Filter</td>
<td>&lt;None&gt;</td>
</tr>
<tr>
<td>MLPP Precedence</td>
<td>Default</td>
</tr>
<tr>
<td>Resource Priority Namespace Network Domain</td>
<td>&lt;None&gt;</td>
</tr>
<tr>
<td>Route Class</td>
<td>Default</td>
</tr>
<tr>
<td>Gateway/Route List</td>
<td>SFB-CUCM_Route List</td>
</tr>
<tr>
<td>Route Option</td>
<td></td>
</tr>
<tr>
<td>Call Classification</td>
<td>OnNet</td>
</tr>
<tr>
<td>External Call Control Profile</td>
<td>&lt;None&gt;</td>
</tr>
<tr>
<td>Allow Device Override</td>
<td>No</td>
</tr>
<tr>
<td>Provide Outside Dial Tone</td>
<td>No</td>
</tr>
<tr>
<td>Allow Overlap Sending</td>
<td>No</td>
</tr>
<tr>
<td>Urgent Priority</td>
<td>No</td>
</tr>
<tr>
<td>Require Forced Authorization Code</td>
<td>No</td>
</tr>
<tr>
<td>Authorization Level</td>
<td></td>
</tr>
<tr>
<td>Require Client Matter Code</td>
<td>No</td>
</tr>
</tbody>
</table>
Cisco Unified Communications Manager Route Pattern to invoke Jabber client with Remote Destination configured as Skype for Business Extensions

Set Route Pattern* = \\+8XXX. This is used to route to the Skype for Business when using the Extend and Connect functionality in this test

Set Description = this text is used to identify this Route Pattern

Set Gateway/Route List* = SFB-CUCM_Route List. This is used for the test

Uncheck Provide Outside Dial Tone

Set Calling Line ID Presentation= Default

Set Calling Name Presentation= Default

Set Connected Line ID Presentation*= Default

Set Calling Name Presentation*= Default

Discard Digits = PreDot
All other values are default
Cisco Unified Communications Manager Route Pattern to Skype for Business Call Park range

The Skype for Business Call Park range configured is 100-199. The following route pattern “1XX” is therefore configured to enable a parked call to be retrieved from Cisco UCM.

![Route Pattern Configuration](image)
Route Pattern Configuration for 1XX (Continued)

- **Calling Party Transformations**
  - Use Calling Party's External Phone Number Mask
  - Calling Party Transform Mask
  - Prefix Digits (Outgoing Calls)
  - Calling Line ID Presentation
  - Calling Name Presentation
  - Calling Party Number Type
  - Calling Party Numbering Plan

- **Connected Party Transformations**
  - Connected Line ID Presentation
  - Connected Name Presentation

- **Called Party Transformations**
  - Discard Digits
  - Called Party Transform Mask
  - Prefix Digits (Outgoing Calls)
  - Called Party Number Type
  - Called Party Numbering Plan

- **ISDN Network-Specific Facilities Information Element**
  - Network Service Protocol
  - Network Identification Code
  - Network Service
  - Service Parameter Name
  - Service Parameter Value
A route pattern 7000 (which is the voice mail pilot), is configured to reach Unity Connection Voice Mail.

<table>
<thead>
<tr>
<th>Pattern Definition</th>
</tr>
</thead>
<tbody>
<tr>
<td>Route Pattern</td>
</tr>
<tr>
<td>Route Partition</td>
</tr>
<tr>
<td>Description</td>
</tr>
<tr>
<td>Numbering Plan</td>
</tr>
<tr>
<td>Route Filter</td>
</tr>
<tr>
<td>MLPP Precedence</td>
</tr>
<tr>
<td>Apply Call Blocking Percentage</td>
</tr>
<tr>
<td>Resource Priority Namespace Network Domain</td>
</tr>
<tr>
<td>Route Class</td>
</tr>
<tr>
<td>Gateway/Route List</td>
</tr>
<tr>
<td>Route Option</td>
</tr>
<tr>
<td>Call Classification</td>
</tr>
<tr>
<td>External Call Control Profile</td>
</tr>
<tr>
<td>Allow Device Override</td>
</tr>
<tr>
<td>Require Forced Authorization Code</td>
</tr>
<tr>
<td>Authorization Level</td>
</tr>
<tr>
<td>Require Client Caller ID</td>
</tr>
</tbody>
</table>

### Calling Party Transformations
- Use Calling Party's External Phone Number Mask
- Calling Party Transform Mask
- Prefix Digits (Outgoing Calls)
Cisco UCM Extent and Connect

Extend and Connect is a feature that allows administrators to rapidly deploy UC Computer Telephony Integration (CTI) applications which interoperate with any endpoint. With Extend and Connect, users can leverage the benefits of UC applications from any location using any device. This feature also allows Interoperability between newer UC solutions and legacy systems, so customers can migrate to newer UC Solutions over time as existing hardware is deprecated.

Cisco UCM UC service Configuration

Navigation Path: User Management ➔ User setting ➔ UC Service

Add New

Select Service Type as CTI

Set Name = CTI_SRV

Set Host Name/IP Address* = 10.80.10.2; this is the Cisco UCM publisher IP.
In the same manner, a UC Service is configured for the subscriber also.

A UC service of Type IM and Presence is configured with the IP of the Presence server.
### Cisco UCM service Profile Configuration

**Navigation:** User Management → User setting → Service Profile

<table>
<thead>
<tr>
<th>Service Profile Configuration</th>
<th>Related Links:</th>
<th>Back To Find/List</th>
</tr>
</thead>
</table>

#### Service Profile Information

- **Name:** jiber_SVC_profile
- **Description:**
- **Make this the default service profile for the system:**

#### Voicemail Profile

- **Primary:** <None>
- **Secondary:** <None>
- **Tertiary:** <None>
- **Credentials source for voicemail service:** Not set

#### MailStore Profile

- **Primary:** <None>
- **Secondary:** <None>
- **Tertiary:** <None>
- **Inbox Folder:** INBOX
- **Trash Folder:** Deleted Items
- **Polling Interval (in seconds):** 60
- **Allow dual folder mode:**
Cisco UCM service profile Configuration (Continued)

<table>
<thead>
<tr>
<th>Service Profile Configuration</th>
<th>Related Links: Back To Find/List</th>
</tr>
</thead>
<tbody>
<tr>
<td><img src="image" alt="Screen capture" /></td>
<td></td>
</tr>
</tbody>
</table>

**Conferencing Profile**

- **Primary**: <None>
- **Secondary**: <None>
- **Tertiary**: <None>
- **Server Certificate Verification**: Any
- **Credentials source for web conference service**: Not set

**Directory Profile**

- **Use UPS for Contact Resolution**
- **Use Logged On User Credential**

**Username**: administrator
**Password**: ********

**Search Base 1**

**Search Base 2**

**Search Base 3**

- **Recursive Search on All Search Bases**
  - **Search Timeout (seconds)**: 5

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Cisco Unified CM IM Presence – CCMCIP Profile Configuration

**Navigation Path:** Application → CCMCIP Profile

Set Name *: remotedesk. This is used in this example.

Set Primary CCMCIP Host *: 10.80.10.2. Cisco Publisher IP. This is used in this test.

Set Backup CCMCIP Host *: 10.80.10.3. Cisco Publisher IP. This is used in this test.

Add Users to Profile: user1, user 2 and user3. This is used in this test.
Cisco UCM – SIP trunk to Cisco IM&Presence Trunk Configuration

Navigation Path: Device → Trunk

Set Device Name* = IMPTrunk. This is used for the test.

Set Description = this text is used to identify this Trunk Group.

Set Device Pool* = Default. This is used for the test.

Set Media Resource Group List = MRGL. This is used for the test.

Set Destination Address = 10.80.10.6. This is used in this example.

Set SIP Trunk Security Profile* = Non Secure SIP Trunk Profile.

Set SIP Profile* = Standard SIP Profile.

Set DTMF Signaling Method* = No Preference.

All other values are default.
Cisco Unified Communications Manager SIP Trunk to CUP Configuration (Continued)
Cisco UCM SIP Trunk to CUP Configuration (Continued)

### Inbound Calls

- **Highlighted**: All
- **Connected Line ID Presentation**: Default
- **Connected Name Presentation**: Default
- **Calling Search Space**: < None >
- **AAR Calling Search Space**: < None >
- **Prefix DN**: 

### Incoming Calling Party Settings

If the administrator sets the prefix to Default this indicates call processing will use prefix at the next level setting (DevicePool/Service Parameter). Otherwise, the value configured is used as the prefix unless the field is empty in which case there is no prefix assigned.

<table>
<thead>
<tr>
<th>Number Type</th>
<th>Prefix</th>
<th>Strip Digits</th>
<th>Calling Search Space</th>
<th>Use Device Pool CSS</th>
</tr>
</thead>
<tbody>
<tr>
<td>Incoming Number</td>
<td>Default</td>
<td>0</td>
<td>&lt; None &gt;</td>
<td></td>
</tr>
</tbody>
</table>

### Incoming Called Party Settings

If the administrator sets the prefix to Default this indicates call processing will use prefix at the next level setting (DevicePool/Service Parameter). Otherwise, the value configured is used as the prefix unless the field is empty in which case there is no prefix assigned.

<table>
<thead>
<tr>
<th>Number Type</th>
<th>Prefix</th>
<th>Strip Digits</th>
<th>Calling Search Space</th>
<th>Use Device Pool CSS</th>
</tr>
</thead>
<tbody>
<tr>
<td>Incoming Number</td>
<td>Default</td>
<td>0</td>
<td>&lt; None &gt;</td>
<td></td>
</tr>
</tbody>
</table>

### Connected Party Settings

- **Connected Party Transformation CSS**: < None >
- **Use Device Pool Connected Party Transformation CSS**: ✔

### Outbound Calls

- **Called Party Transformation CSS**: < None >
- **Use Device Pool Called Party Transformation CSS**: ✔
- **Calling Party Transformation CSS**: < None >
- **Use Device Pool Calling Party Transformation CSS**: ✔
- **Calling Party Selection**: Originator
- **Calling Line ID Presentation**: Default
- **Calling Name Presentation**: Default
- **Calling and Connected Party Info Format**: Deliver DN only in connected party
- **Redirecting Diversion Header Delivery - Outbound**: ✔
- **Redirecting Party Transformation CSS**: < None >
- **Use Device Pool Redirecting Party Transformation CSS**: ✔

### Caller Information

- **Caller ID DN**: 
- **Caller Name**: 
- **Maintain Original Caller ID DN and Caller Name in Identity Headers**: ✔

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Cisco UCM SIP Trunk to CUP Configuration (Continued)
Cisco UCM end user configuration

Add user to Cisco UCM

**Navigation:** User Management ➔ End user

Set User ID* = user1. This is used for the test.

Set Last Name = user1. This is used for the test.

Check Home Cluster.

Click the Device Association

Select CTI1 from User Device Association screen
Cisco UCM end user Configuration (Continued)

<table>
<thead>
<tr>
<th>End User Configuration</th>
<th>Related Links: Back to Find List Users ▼ Go</th>
</tr>
</thead>
<tbody>
<tr>
<td>Digest Credentials</td>
<td></td>
</tr>
<tr>
<td>Confirm Digest Credentials</td>
<td></td>
</tr>
<tr>
<td>User Profile</td>
<td>Use System Default (Standard (Factory Default)) ▼ View Details</td>
</tr>
</tbody>
</table>

**Service Settings**

- [x] Home Cluster
- [x] Enable User for Unified CM IM and Presence (Configure IM and Presence in the associated UC Service Profile)
- Include meeting information in presence (requires Exchange Presence Gateway to be configured on CUCM IM and Presence server)
- Presence Viewer for User
- UC Service Profile: jobber_SVC_profile ▼ view Details

**Device Information**

<table>
<thead>
<tr>
<th>Controlled Devices</th>
<th>CT1</th>
</tr>
</thead>
</table>

Available Profiles

CT1 Controlled Device Profiles

**Device Association**

Line Appearance Association for Presence
Check Allow Control of Device from CTI
Select the Primary Extension for this user. 5007 is used for this example.
Check Enable Mobility

Add the following permissions for Standard Users:
- Standard CCM End-Users
- Standard CTI Enabled
- Standard CCMUSER Administration
Remote Destination Configuration

**Navigation:** Device ➔ Remote Destination

Add New

Set name = Jabber RD. This is used for the test

Set Destination Number*= +8004. This is used for the test. [8004 is a Skype for Business extension]

Check Enable Extend and Connect.

Set CTI Remote Device = CTI1

---

The CTI Remote Device configuration is updated with the remote destination:

---

Two Remote Destinations were configured for this test:
Cisco UCM CTI Remote Device Configuration

**Navigation:** Device→Phone

Add New.

Select Phone Type * = CTI Remote Device

The CTI Remote Device type represents the user’s remote device(s).

Select the desired Owner User ID. user1 is used in this test.

Set Device Pool: G711 Preferred

Save.
Add a DN to this device.

DN 7009 was configured for this test.
Cisco UCM CTI Remote Device DN Configuration

**Directory Number Configuration**

<table>
<thead>
<tr>
<th>Directory Number</th>
<th>Route Partition</th>
<th>Description</th>
<th>Alerting Name</th>
<th>ASCII Alerting Name</th>
<th>External Call Control Profile</th>
<th>Allow Control of Device from CTI</th>
<th>Associated Devices</th>
</tr>
</thead>
<tbody>
<tr>
<td>7009</td>
<td>&lt; None &gt;</td>
<td></td>
<td>Jabber_Lynch004</td>
<td>Jabber_Lynch004</td>
<td>&lt; None &gt;</td>
<td>✔</td>
<td>CT1</td>
</tr>
</tbody>
</table>

**Directory Number Settings**

- **Voice Mail Profile**: < None >
- **Calling Search Space**: < None >
- **BLF Presence Group**: Standard Presence group
- **User Hold MOH Audio Source**: < None >
- **Network Hold MOH Audio Source**: < None >
- **Reject Anonymous Calls**: ✔

---

**Related Links**: Configure Device (CTI)
Cisco Unity Connection

Cisco Unity Connection Telephony Integration – Add Phone System

**Navigation:** Telephony Integrations → Phone system

Add New

Set Phone System Name* = SFB_TLS. This Name used for this test

Check Use Same Port for Enabling and Disabling MWIs

---

### Cisco Unity Connection Telephony Integration – Add Port Group

**Navigation:** Telephony Integration → Port Group or from previous Screen, Related Links “Add Port Group”

Go

Set Phone System = SFB_TLS

Set Create From – Port group Type = SIP

Set Display Name* = SFB_TLS-1. This Name used for this example.

Set Ipv4 Address or Host Name = 10.80.10.2 [This is the Cisco UCM publisher IP]
Check Register with SIP server

Set Security Profile = 5061/TLS

SIP Certificate = Secure SIP Integration with CUCMv10.5 (which is the display name of SIP Certificate generated in CUC under Security)

Set Security Mode = Encrypted

Check Secure RTP

Check Enable Message Waiting Indicators

Click Save.
Cisco Unity Connection Telephony Integration – Add Ports

Set Number of Ports = 10
Set Phone System = SFB_TLS
Set Port Group = SFB_TLS-1
Set Server = clus20unity.skypelabsj.local (which is the FQDN of Cisco Unity Server)
Cisco Unity Connection User Configuration

Navigation: Cisco Unity Connection → Users → Users

Set Alias* = 8004 (This is used for the test)
Set First Name = User 1 (This is used to identify the User)
Set Last Name = SFB
Set Extension* = 8004 (This is user’s extension number)
Save.
Set Phone System = SFB_TLS

Cisco Unity Connection User Configuration (Continued)
Cisco Unity Connection User Configuration (Continued)
Cisco Unity Connection User Configuration (Continued)

All values are default.

Similarly, create a user that has a Cisco extension.
### Acronyms

<table>
<thead>
<tr>
<th>Acronym</th>
<th>Definition</th>
</tr>
</thead>
<tbody>
<tr>
<td>CCNR</td>
<td>Call Completion on No Reply</td>
</tr>
<tr>
<td>CFB</td>
<td>Call Forwarding on Busy</td>
</tr>
<tr>
<td>CFNA</td>
<td>Call Forwarding No Answer</td>
</tr>
<tr>
<td>CFU</td>
<td>Call Forwarding Unconditional</td>
</tr>
<tr>
<td>Cisco UCM</td>
<td>Cisco Unified Communications Manager</td>
</tr>
<tr>
<td>CLIP</td>
<td>Calling Line (Number) Identification Presentation</td>
</tr>
<tr>
<td>CLIR</td>
<td>Calling Line (Number) Identification Restriction</td>
</tr>
<tr>
<td>CNIP</td>
<td>Calling Name Identification Presentation</td>
</tr>
<tr>
<td>CNIR</td>
<td>Calling Name Identification Restriction</td>
</tr>
<tr>
<td>COLP</td>
<td>Connected Line (Number) Identification Presentation</td>
</tr>
<tr>
<td>COLR</td>
<td>Connected Line (Number) Identification Restriction</td>
</tr>
<tr>
<td>CONP</td>
<td>Connected Name Identification Presentation</td>
</tr>
<tr>
<td>CONR</td>
<td>Connected Name Identification Restriction</td>
</tr>
<tr>
<td>CT</td>
<td>Call Transfer</td>
</tr>
<tr>
<td>CUP</td>
<td>Cisco Unified Presence</td>
</tr>
<tr>
<td>DNS</td>
<td>Domain Name Server</td>
</tr>
<tr>
<td>EXT</td>
<td>Extension</td>
</tr>
<tr>
<td>FQDN</td>
<td>Fully Qualified Domain Name</td>
</tr>
<tr>
<td>MRGL</td>
<td>Media Resource Group List</td>
</tr>
<tr>
<td>MTP</td>
<td>Media Termination Point</td>
</tr>
<tr>
<td>MWI</td>
<td>Message Waiting Indicator</td>
</tr>
<tr>
<td>PBX</td>
<td>Private Branch Exchange</td>
</tr>
<tr>
<td>PSTN</td>
<td>Public Switched Telephone Network</td>
</tr>
<tr>
<td>RTP</td>
<td>Real Time Protocol</td>
</tr>
<tr>
<td>SCCP</td>
<td>Skinny Client Control Protocol</td>
</tr>
<tr>
<td>SFB</td>
<td>Skype for Business</td>
</tr>
<tr>
<td>SIP</td>
<td>Session Initiated Protocol</td>
</tr>
<tr>
<td>UDP</td>
<td>Uniform Dial Plan</td>
</tr>
<tr>
<td>VM</td>
<td>Voice Mail</td>
</tr>
</tbody>
</table>